

Network Working Group
Internet-Draft
Intended status: Standards Track
Expires: October 28, 2021

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April 26, 2021

Metrics and Methods for One-way IP Capacity
draft-ietf-ippm-capacity-metric-method-10

Abstract

This memo revisits the problem of Network Capacity metrics first examined in [RFC 5136](#). The memo specifies a more practical Maximum IP-Layer Capacity metric definition catering for measurement purposes, and outlines the corresponding methods of measurement.

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1. Introduction

The IETF's efforts to define Network and Bulk Transport Capacity have been chartered and progressed for over twenty years. Over that time, the performance community has seen development of Informative definitions in [\[RFC3148\]](#) for Framework for Bulk Transport Capacity (BTC), [RFC 5136](#) for Network Capacity and Maximum IP-Layer Capacity, and the Experimental metric definitions and methods in [\[RFC8337\]](#), Model-Based Metrics for BTC.

This memo revisits the problem of Network Capacity metrics examined first in [\[RFC3148\]](#) and later in [\[RFC5136\]](#). Maximum IP-Layer Capacity and [\[RFC3148\]](#) Bulk Transfer Capacity (goodput) are different metrics. Maximum IP-Layer Capacity is like the theoretical goal for goodput. There are many metrics in [\[RFC5136\]](#), such as Available Capacity. Measurements depend on the network path under test and the use case. Here, the main use case is to assess the maximum capacity of the access network, with specific performance criteria used in the measurement.

This memo recognizes the importance of a definition of a Maximum IP-Layer Capacity Metric at a time when access speeds have increased dramatically; a definition that is both practical and effective for the performance community's needs, including Internet users. The metric definition is intended to use Active Methods of Measurement [\[RFC7799\]](#), and a method of measurement is included.

The most direct active measurement of IP-Layer Capacity would use IP packets, but in practice a transport header is needed to traverse address and port translators. UDP offers the most direct assessment possibility, and in the [\[copycat\]](#) measurement study to investigate whether UDP is viable as a general Internet transport protocol, the authors found that a high percentage of paths tested support UDP transport. A number of liaisons have been exchanged on this topic [\[LS-SG12-A\]](#) [\[LS-SG12-B\]](#), discussing the laboratory and field tests that support the UDP-based approach to IP-Layer Capacity measurement.

This memo also recognizes the many updates to the IP Performance Metrics Framework [\[RFC2330\]](#) published over twenty years, and makes use of [\[RFC7312\]](#) for Advanced Stream and Sampling Framework, and [\[RFC8468\]](#) with IPv4, IPv6, and IPv4-IPv6 Coexistence Updates.

[Appendix A](#) describes the load rate adjustment algorithm in pseudo-code. [Appendix B](#) discusses the algorithm's compliance with [\[RFC8085\]](#).

1.1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [BCP 14](#)[\[RFC2119\]](#) [\[RFC8174\]](#) when, and only when, they appear in all capitals, as shown here.

2. Scope, Goals, and Applicability

The scope of this memo is to define a metric and corresponding method to unambiguously perform Active measurements of Maximum IP-Layer Capacity, along with related metrics and methods.

Another goal is to harmonize the specified metric and method across the industry, and this memo is the vehicle that captures IETF consensus, possibly resulting in changes to the specifications of other Standards Development Organizations (SDO) (through each SDO's normal contribution process, or through liaison exchange).

A local goal is to aid efficient test procedures where possible, and to recommend reporting with additional interpretation of the results. Fostering the development of protocol support for this metric and method of measurement is also a goal of this memo (all active testing protocols currently defined by the IPPM WG are UDP-based, meeting a key requirement of these methods). The supporting protocol development to measure this metric according to the specified method is a key future contribution to Internet measurement.

The load rate adjustment algorithm's scope is limited to helping determine the Maximum IP-Layer Capacity in the context of an infrequent, diagnostic, short term measurement. It is RECOMMENDED to discontinue non-measurement traffic that shares a subscriber's dedicated resources while testing: measurements may not be accurate and throughput of competing elastic traffic may be greatly reduced.

The primary application of the metric and method of measurement described here is the same as in [Section 2 of \[RFC7497\]](#) where:

- o The access portion of the network is the focus of this problem statement. The user typically subscribes to a service with bidirectional access partly described by rates in bits per second.

In addition, the use of the load rate adjustment algorithm described in [section 8.1](#) has the following additional applicability limitations:

- MUST only be used in the application of diagnostic and operations measurements as described in this memo
- MUST only be used in circumstances consistent with [Section 10](#), Security Considerations
- If a network operator is certain of the access capacity to be validated, then testing MAY start with a fixed rate test at the access capacity and avoid activating the load adjustment algorithm. However, the stimulus for a diagnostic test (such as a subscriber request) strongly implies that there is no certainty and the load adjustment algorithm will be needed.

Further, the metric and method of measurement are intended for use where specific exact path information is unknown within a range of possible values:

- the subscriber's exact Maximum IP-Layer Capacity is unknown (which is sometimes the case; service rates can be increased due to upgrades without a subscriber's request, or to provide a surplus to compensate for possible underestimates of TCP-based testing.
- the size of the access bottleneck buffer is unknown.

Finally, the measurement system's load rate adjustment algorithm SHALL NOT be provided with the exact capacity value to be validated a priori. This restriction fosters a fair result, and removes an opportunity for bad actors to operate with knowledge of the "right answer".

3. Motivation

As with any problem that has been worked for many years in various SDOs without any special attempts at coordination, various solutions for metrics and methods have emerged.

There are five factors that have changed (or begun to change) in the 2013-2019 time frame, and the presence of any one of them on the path requires features in the measurement design to account for the changes:

1. Internet access is no longer the bottleneck for many users.
2. Both transfer rate and latency are important to user's satisfaction.
3. UDP's growing role in Transport, in areas where TCP once dominated.

4. Content and applications are moving physically closer to users.
5. There is less emphasis on ISP gateway measurements, possibly due to less traffic crossing ISP gateways in future.

4. General Parameters and Definitions

This section lists the REQUIRED input factors to specify a Sender or Receiver metric.

- o Src, the address of a host (such as the globally routable IP address).
- o Dst, the address of a host (such as the globally routable IP address).
- o MaxHops, the limit on the number of Hops a specific packet may visit as it traverses from the host at Src to the host at Dst (implemented in the TTL or Hop Limit).
- o T0, the time at the start of measurement interval, when packets are first transmitted from the Source.
- o I, the nominal duration of a measurement interval at the destination (default 10 sec)
- o dt, the nominal duration of m equal sub-intervals in I at the destination (default 1 sec)
- o dtn, the beginning boundary of a specific sub-interval, n, one of m sub-intervals in I
- o FT, the feedback time interval between status feedback messages communicating measurement results, sent from the receiver to control the sender. The results are evaluated throughout the test to determine how to adjust the current offered load rate at the sender (default 50ms)
- o Tmax, a maximum waiting time for test packets to arrive at the destination, set sufficiently long to disambiguate packets with long delays from packets that are discarded (lost), such that the distribution of one-way delay is not truncated.
- o F, the number of different flows synthesized by the method (default 1 flow)
- o flow, the stream of packets with the same n-tuple of designated header fields that (when held constant) result in identical

treatment in a multi-path decision (such as the decision taken in load balancing). Note: The IPv6 flow label MAY be included in the flow definition when routers have complied with [\[RFC6438\]](#) guidelines.

- o Type-P, the complete description of the test packets for which this assessment applies (including the flow-defining fields). Note that the UDP transport layer is one requirement for test packets specified below. Type-P is a parallel concept to "population of interest" defined in clause 6.1.1 of[Y.1540].
- o PM, a list of fundamental metrics, such as loss, delay, and reordering, and corresponding target performance threshold. At least one fundamental metric and target performance threshold MUST be supplied (such as One-way IP Packet Loss [\[RFC7680\]](#) equal to zero).

A non-Parameter which is required for several metrics is defined below:

- o T, the host time of the *first* test packet's *arrival* as measured at the destination Measurement Point, or MP(Dst). There may be other packets sent between Source and Destination hosts that are excluded, so this is the time of arrival of the first packet used for measurement of the metric.

Note that time stamp format and resolution, sequence numbers, etc. will be established by the chosen test protocol standard or implementation.

5. IP-Layer Capacity Singleton Metric Definitions

This section sets requirements for the singleton metric that supports the Maximum IP-Layer Capacity Metric definition in [Section 6](#).

[5.1.](#) Formal Name

Type-P-One-way-IP-Capacity, or informally called IP-Layer Capacity.

Note that Type-P depends on the chosen method.

[5.2.](#) Parameters

This section lists the REQUIRED input factors to specify the metric, beyond those listed in [Section 4](#).

No additional Parameters are needed.

5.3. Metric Definitions

This section defines the REQUIRED aspects of the measurable IP-Layer Capacity metric (unless otherwise indicated) for measurements between specified Source and Destination hosts:

Define the IP-Layer Capacity, $C(T,dt,PM)$, to be the number of IP-Layer bits (including header and data fields) in packets that can be transmitted from the Src host and correctly received by the Dst host during one contiguous sub-interval, dt in length. The IP-Layer Capacity depends on the Src and Dst hosts, the host addresses, and the path between the hosts.

The number of these IP-Layer bits is designated $n0[dt_n,dt_{n+1}]$ for a specific dt .

When the packet size is known and of fixed size, the packet count during a single sub-interval dt multiplied by the total bits in IP header and data fields is equal to $n0[dt_n,dt_{n+1}]$.

Anticipating a Sample of Singletons, the number of sub-intervals with duration dt MUST be set to a natural number m , so that $T+I = T + m*dt$ with $dt_{n+1} - dt_n = dt$ for $1 \leq n \leq m$.

Parameter PM represents other performance metrics [see [section 5.4](#) below]; their measurement results SHALL be collected during measurement of IP-Layer Capacity and associated with the corresponding dt_n for further evaluation and reporting. Users SHALL specify the parameter T_{max} as required by each metric's reference definition.

Mathematically, this definition is represented as (for each n):

$$C(T,dt,PM) = \frac{(n0[dt_n,dt_{n+1}])}{dt}$$

Equation for IP-Layer Capacity

and:

- o $n0$ is the total number of IP-Layer header and payload bits that can be transmitted in standard-formed packets [[RFC8468](#)] from the Src host and correctly received by the Dst host during one contiguous sub-interval, dt in length, during the interval $[T, T+I]$,

- o $C(T, dt, PM)$ the IP-Layer Capacity, corresponds to the value of n_0 measured in any sub-interval beginning at dt_n , divided by the length of sub-interval, dt .
- o PM represents other performance metrics [see [section 5.4](#) below]; their measurement results SHALL be collected during measurement of IP-Layer Capacity and associated with the corresponding dt_n for further evaluation and reporting.
- o all sub-intervals MUST be of equal duration. Choosing dt as non-overlapping consecutive time intervals allows for a simple implementation.
- o The bit rate of the physical interface of the measurement devices MUST be higher than the smallest of the links on the path whose $C(T, I, PM)$ is to be measured (the bottleneck link).

Measurements according to these definitions SHALL use the UDP transport layer. Standard-formed packets are specified in [Section 5 of \[RFC8468\]](#). The measurement SHOULD use a randomized Source port or equivalent technique, and SHOULD send responses from the Source address matching the test packet destination address.

Some compression affects on measurement are discussed in [Section 6 of \[RFC8468\]](#).

5.4. Related Round-Trip Delay and One-way Loss Definitions

$RTD[dt_n, dt_n+1]$ is defined as a Sample of the [\[RFC2681\]](#) Round-trip Delay between the Src host and the Dst host over the interval $[T, T+I]$ (that contains equal non-overlapping intervals of dt). The "reasonable period of time" in [\[RFC2681\]](#) is the parameter T_{max} in this memo. The statistics used to summarize $RTD[dt_n, dt_n+1]$ MAY include the minimum, maximum, median, and mean, and the range = (maximum - minimum) is referred to below in [Section 8.1](#) for load adjustment purposes.

$OWL[dt_n, dt_n+1]$ is defined as a Sample of the [\[RFC7680\]](#) One-way Loss between the Src host and the Dst host over the interval $[T, T+I]$ (that contains equal non-overlapping intervals of dt). The statistics used to summarize $OWL[dt_n, dt_n+1]$ MAY include the lost packet count and the lost packet ratio.

Other metrics MAY be measured: one-way reordering, duplication, and delay variation.

[5.5.](#) Discussion

See the corresponding section for Maximum IP-Layer Capacity.

[5.6.](#) Reporting the Metric

The IP-Layer Capacity SHOULD be reported with at least single Megabit resolution, in units of Megabits per second (Mbps), (which is 1,000,000 bits per second to avoid any confusion).

The related One-way Loss metric and Round Trip Delay measurements for the same Singleton SHALL be reported, also with meaningful resolution for the values measured.

Individual Capacity measurements MAY be reported in a manner consistent with the Maximum IP-Layer Capacity, see [Section 9](#).

[6.](#) Maximum IP-Layer Capacity Metric Definitions (Statistic)

This section sets requirements for the following components to support the Maximum IP-Layer Capacity Metric.

[6.1.](#) Formal Name

Type-P-One-way-Max-IP-Capacity, or informally called Maximum IP-Layer Capacity.

Note that Type-P depends on the chosen method.

[6.2.](#) Parameters

This section lists the REQUIRED input factors to specify the metric, beyond those listed in [Section 4](#).

No additional Parameters or definitions are needed.

[6.3.](#) Metric Definitions

This section defines the REQUIRED aspects of the Maximum IP-Layer Capacity metric (unless otherwise indicated) for measurements between specified Source and Destination hosts:

Define the Maximum IP-Layer Capacity, $\text{Maximum_C}(T, I, PM)$, to be the maximum number of IP-Layer bits $n_0[dt_n, dt_n+1]$ divided by dt that can be transmitted in packets from the Src host and correctly received by the Dst host, over all dt length intervals in $[T, T+I]$, and meeting the PM criteria. Equivalently the Maximum of a Sample of size m of

$C(T,I,PM)$ collected during the interval $[T, T+I]$ and meeting the PM criteria.

The number of sub-intervals with duration dt MUST be set to a natural number m , so that $T+I = T + m*dt$ with $dt_{n+1} - dt_n = dt$ for $1 \leq n \leq m$.

Parameter PM represents the other performance metrics (see [Section 6.4](#) below) and their measurement results for the Maximum IP-Layer Capacity. At least one target performance threshold (PM criterion) MUST be defined. If more than one metric and target performance threshold are defined, then the sub-interval with maximum number of bits transmitted MUST meet all the target performance thresholds. Users SHALL specify the parameter Tmax as required by each metric's reference definition.

Mathematically, this definition can be represented as:

$$\text{Maximum_C}(T, I, PM) = \frac{\max_{[T, T+I]} (n0[dt_n, dt_{n+1}])}{dt}$$

where:

T											$T+I$
$dt_n=1$	2	3	4	5	6	7	8	9	10	$n+1$	
$n=m$											

Equation for Maximum Capacity

and:

- o $n0$ is the total number of IP-Layer header and payload bits that can be transmitted in standard-formed packets from the Src host and correctly received by the Dst host during one contiguous sub-interval, dt in length, during the interval $[T, T+I]$,
- o $\text{Maximum_C}(T, I, PM)$ the Maximum IP-Layer Capacity, corresponds to the maximum value of $n0$ measured in any sub-interval beginning at dt_n , divided by the constant length of all sub-intervals, dt .
- o PM represents the other performance metrics (see [Section 5.4](#)) and their measurement results for the Maximum IP-Layer Capacity. At least one target performance threshold (PM criterion) MUST be defined.

- o all sub-intervals MUST be of equal duration. Choosing dt as non-overlapping consecutive time intervals allows for a simple implementation.
- o The bit rate of the physical interface of the measurement systems MUST be higher than the smallest of the links on the path whose $Maximum_C(T,I,PM)$ is to be measured (the bottleneck link).

In this definition, the m sub-intervals can be viewed as trials when the Src host varies the transmitted packet rate, searching for the maximum n_0 that meets the PM criteria measured at the Dst host in a test of duration, I . When the transmitted packet rate is held constant at the Src host, the m sub-intervals may also be viewed as trials to evaluate the stability of n_0 and metric(s) in the PM list over all dt -length intervals in I .

Measurements according to these definitions SHALL use the UDP transport layer.

6.4. Related Round-Trip Delay and One-way Loss Definitions

$RTD[dt_n, dt_{n+1}]$ and $OWL[dt_n, dt_{n+1}]$ are defined in [Section 5.4](#). Here, the test intervals are increased to match the capacity Samples, $RTD[T, I]$ and $OWL[T, I]$.

The interval dt_n, dt_{n+1} where $Maximum_C[T, I, PM]$ occurs is the reporting sub-interval within $RTD[T, I]$ and $OWL[T, I]$.

Other metrics MAY be measured: one-way reordering, duplication, and delay variation.

6.5. Discussion

If traffic conditioning (e.g., shaping, policing) applies along a path for which $Maximum_C(T, I, PM)$ is to be determined, different values for dt SHOULD be picked and measurements be executed during multiple intervals $[T, T+I]$. Each duration dt SHOULD be chosen so that it is an integer multiple of increasing values k times serialization delay of a path MTU at the physical interface speed where traffic conditioning is expected. This should avoid taking configured burst tolerance singletons as a valid $Maximum_C(T, I, PM)$ result.

A $Maximum_C(T, I, PM)$ without any indication of bottleneck congestion, be that an increasing latency, packet loss or ECN marks during a measurement interval I , is likely to underestimate $Maximum_C(T, I, PM)$.

6.6. Reporting the Metric

The IP-Layer Capacity SHOULD be reported with at least single Megabit resolution, in units of Megabits per second (Mbps) (which is 1,000,000 bits per second to avoid any confusion).

The related One-way Loss metric and Round Trip Delay measurements for the same Singleton SHALL be reported, also with meaningful resolution for the values measured.

When there are demonstrated and repeatable Capacity modes in the Sample, then the Maximum IP-Layer Capacity SHALL be reported for each mode, along with the relative time from the beginning of the stream that the mode was observed to be present. Bimodal Maximum IP-Layer Capacities have been observed with some services, sometimes called a "turbo mode" intending to deliver short transfers more quickly, or reduce the initial buffering time for some video streams. Note that modes lasting less than dt duration will not be detected.

Some transmission technologies have multiple methods of operation that may be activated when channel conditions degrade or improve, and these transmission methods may determine the Maximum IP-Layer Capacity. Examples include line-of-sight microwave modulator constellations, or cellular modem technologies where the changes may be initiated by a user moving from one coverage area to another. Operation in the different transmission methods may be observed over time, but the modes of Maximum IP-Layer Capacity will not be activated deterministically as with the "turbo mode" described in the paragraph above.

7. IP-Layer Sender Bit Rate Singleton Metric Definitions

This section sets requirements for the following components to support the IP-Layer Sender Bitrate Metric. This metric helps to check that the sender actually generated the desired rates during a test, and measurement takes place at the Src host to network path interface (or as close as practical within the Src host). It is not a metric for path performance.

7.1. Formal Name

Type-P-IP-Sender-Bit-Rate, or informally called IP-Layer Sender Bitrate.

Note that Type-P depends on the chosen method.

7.2. Parameters

This section lists the REQUIRED input factors to specify the metric, beyond those listed in [Section 4](#).

- o S, the duration of the measurement interval at the Source
- o st, the nominal duration of N sub-intervals in S (default st = 0.05 seconds)
- o stn, the beginning boundary of a specific sub-interval, n, one of N sub-intervals in S

S SHALL be longer than I, primarily to account for on-demand activation of the path, or any preamble to testing required, and the delay of the path.

st SHOULD be much smaller than the sub-interval dt and on the same order as FT, otherwise the rate measurement will include many rate adjustments and include more time smoothing, thus missing the Maximum IP-Layer Capacity. The st parameter does not have relevance when the Source is transmitting at a fixed rate throughout S.

7.3. Metric Definition

This section defines the REQUIRED aspects of the IP-Layer Sender Bitrate metric (unless otherwise indicated) for measurements at the specified Source on packets addressed for the intended Destination host and matching the required Type-P:

Define the IP-Layer Sender Bit Rate, $B(S, st)$, to be the number of IP-Layer bits (including header and data fields) that are transmitted from the Source with address pair Src and Dst during one contiguous sub-interval, st, during the test interval S (where S SHALL be longer than I), and where the fixed-size packet count during that single sub-interval st also provides the number of IP-Layer bits in any interval, $[stn, stn+1]$.

Measurements according to these definitions SHALL use the UDP transport layer. Any feedback from Dst host to Src host received by Src host during an interval $[stn, stn+1]$ SHOULD NOT result in an adaptation of the Src host traffic conditioning during this interval (rate adjustment occurs on st interval boundaries).

7.4. Discussion

Both the Sender and Receiver or (Source and Destination) bit rates SHOULD be assessed as part of an IP-Layer Capacity measurement. Otherwise, an unexpected sending rate limitation could produce an erroneous Maximum IP-Layer Capacity measurement.

7.5. Reporting the Metric

The IP-Layer Sender Bit Rate SHALL be reported with meaningful resolution, in units of Megabits per second (which is 1,000,000 bits per second to avoid any confusion).

Individual IP-Layer Sender Bit Rate measurements are discussed further in [Section 9](#).

8. Method of Measurement

The architecture of the method REQUIRES two cooperating hosts operating in the roles of Src (test packet sender) and Dst (receiver), with a measured path and return path between them.

The duration of a test, parameter I, MUST be constrained in a production network, since this is an active test method and it will likely cause congestion on the Src to Dst host path during a test.

8.1. Load Rate Adjustment Algorithm

The algorithm described in this section MUST NOT be used as a general Congestion Control Algorithm (CCA). As stated in the Scope [Section 2](#), the load rate adjustment algorithm's goal is to help determine the Maximum IP-Layer Capacity in the context of an infrequent, diagnostic, short term measurement. There is a tradeoff between test duration (also the test data volume) and algorithm aggressiveness (speed of ramp-up and down to the Maximum IP-Layer Capacity). The parameter values chosen below strike a well-tested balance among these factors.

A table SHALL be pre-built defining all the offered load rates that will be supported (R1 through Rn, in ascending order, corresponding to indexed rows in the table). It is RECOMMENDED that rates begin with 0.5 Mbps at index zero, use 1 Mbps at index one, and then continue in 1 Mbps increments to 1 Gbps. Above 1 Gbps, and up to 10 Gbps, it is RECOMMENDED that 100 Mbps increments be used. Above 10 Gbps, increments of 1 Gbps are RECOMMENDED. A higher initial IP-Layer Sender Bitrate might be configured when the test operator is certain that the Maximum IP-Layer Capacity is well-above the initial

IP-Layer Sender Bitrate and factors such as test duration and total test traffic play an important role.

Each rate is defined as datagrams of size ss , sent as a burst of count cc , each time interval tt (default for tt is 1ms, a likely system tick-interval). While it is advantageous to use datagrams of as large a size as possible, it may be prudent to use a slightly smaller maximum that allows for secondary protocol headers and/or tunneling without resulting in IP-Layer fragmentation. Selection of a new rate is indicated by a calculation on the current row, Rx . For example:

" $Rx+1$ ": the sender uses the next higher rate in the table.

" $Rx-10$ ": the sender uses the rate 10 rows lower in the table.

At the beginning of a test, the sender begins sending at rate $R1$ and the receiver starts a feedback timer of duration FT (while awaiting inbound datagrams). As datagrams are received they are checked for sequence number anomalies (loss, out-of-order, duplication, etc.) and the delay range is measured (one-way or round-trip). This information is accumulated until the feedback timer FT expires and a status feedback message is sent from the receiver back to the sender, to communicate this information. The accumulated statistics are then reset by the receiver for the next feedback interval. As feedback messages are received back at the sender, they are evaluated to determine how to adjust the current offered load rate (Rx).

If the feedback indicates that no sequence number anomalies were detected AND the delay range was below the lower threshold, the offered load rate is increased. If congestion has not been confirmed up to this point, the offered load rate is increased by more than one rate (e.g., $Rx+10$). This allows the offered load to quickly reach a near-maximum rate. Conversely, if congestion has been previously confirmed, the offered load rate is only increased by one ($Rx+1$). However, if a rate threshold between high and very high sending rates (such as 1 Gbps) is exceeded, the offered load rate is only increased by one ($Rx+1$) above the rate threshold in any congestion state.

If the feedback indicates that sequence number anomalies were detected OR the delay range was above the upper threshold, the offered load rate is decreased. The RECOMMENDED values are 0 for sequence number gaps and 30 ms for lower and 90 ms for upper delay thresholds, respectively. Also, if congestion is now confirmed for the first time by the current feedback message being processed, then the offered load rate is decreased by more than one rate (e.g., $Rx-30$). This one-time reduction is intended to compensate for the fast

initial ramp-up. In all other cases, the offered load rate is only decreased by one (Rx-1).

If the feedback indicates that there were no sequence number anomalies AND the delay range was above the lower threshold, but below the upper threshold, the offered load rate is not changed. This allows time for recent changes in the offered load rate to stabilize, and the feedback to represent current conditions more accurately.

Lastly, the method for inferring congestion is that there were sequence number anomalies AND/OR the delay range was above the upper threshold for two consecutive feedback intervals. The algorithm described above is also illustrated in ITU-T Rec. Y.1540, 2020 version[Y.1540], in Annex B, and implemented in the Appendix on Load Rate Adjustment Pseudo Code in this memo.

The load rate adjustment algorithm MUST include timers that stop the test when received packet streams cease unexpectedly. The timeout thresholds are provided in the table below, along with values for all other parameters and variables described in this section. Operation of non-obvious parameters appear below:

load packet timeout Operation: The load packet timeout SHALL be reset to the configured value each time a load packet received. If the timeout expires, the receiver SHALL be closed and no further feedback sent.

feedback message timeout Operation: The feedback message timeout SHALL be reset to the configured value each time a feedback message is received. If the timeout expires, the sender SHALL be closed and no further load packets sent.

Parameter	Default	Tested Range or values	Expected Safe Range (not entirely tested, other values NOT RECOMMENDED)
FT, feedback time interval	50ms	20ms, 50ms, 100ms	20ms <= FT <= 250ms Larger values may slow the rate increase and fail to find the max
Feedback message timeout	L*FT, L=20 (1sec with FT=50ms)	L=100 with FT=50ms (5sec)	0.5sec <= L*FT <= 30sec Upper limit for very unreliable test

(stop test)			paths only
+-----+	+-----+	+-----+	+-----+
load packet	1sec	5sec	0.250sec - 30sec
timeout			Upper limit for very
(stop test)			unreliable test paths
			only
+-----+	+-----+	+-----+	+-----+
table index	0.5Mbps	0.5Mbps	when testing <=10Gbps
0			
+-----+	+-----+	+-----+	+-----+
table index	1Mbps	1Mbps	when testing <=10Gbps
1			
+-----+	+-----+	+-----+	+-----+
table index	1Mbps	1Mbps<=rate<=	same as tested
(step) size		1Gbps	
+-----+	+-----+	+-----+	+-----+
table index	100Mbps	1Gbps<=rate<=	same as tested
(step)		10Gbps	
size,			
rate>1Gbps			
+-----+	+-----+	+-----+	+-----+
table index	1Gbps	untested	>10Gbps
(step)			
size,			
rate>10Gbps			
+-----+	+-----+	+-----+	+-----+
ss, UDP	none	<=1222	Recommend max at
payload			largest value that
size, bytes			avoids fragmentation;
			use of too-small
			payload size might
			result in unexpected
			sender limitations.
+-----+	+-----+	+-----+	+-----+
cc, burst	none	1<=cc<= 100	same as tested. Vary
count			cc as needed to
			create the desired
			maximum sending rate.
			Sender buffer size
			may limit cc in
			implementation.
+-----+	+-----+	+-----+	+-----+
tt, burst	100microsec	100microsec,	available range of
interval		1msec	"tick" values (HZ
			param)
+-----+	+-----+	+-----+	+-----+
low delay	30ms	5ms, 30ms	same as tested
range			

threshold				
+-----+	+-----+	+-----+	+-----+	+-----+
high delay	90ms	10ms, 90ms	same as tested	
range				
threshold				
+-----+	+-----+	+-----+	+-----+	+-----+
sequence	0	0, 100	same as tested	
error				
threshold				
+-----+	+-----+	+-----+	+-----+	+-----+
consecutive	2	2	Use values >1 to	
errored			avoid misinterpreting	
status			transient loss	
report				
threshold				
+-----+	+-----+	+-----+	+-----+	+-----+
Fast mode	10	10	2 <= steps <= 30	
increase,				
in table				
index steps				
+-----+	+-----+	+-----+	+-----+	+-----+
Fast mode	3 * Fast	3 * Fast mode	same as tested	
decrease,	mode	increase		
in table	increase			
index steps				
+-----+	+-----+	+-----+	+-----+	+-----+

Parameters for Load Rate Adjustment Algorithm

As a consequence of default parameterization, the Number of table steps in total for rates <10Gbps is 2000 (excluding index 0).

A related sender backoff response to network conditions occurs when one or more status feedback messages fail to arrive at the sender.

If no status feedback messages arrive at the sender for the interval greater than the Lost Status Backoff timeout:

$$\text{UDRT} + (2+w)*\text{FT} = \text{Lost Status Backoff timeout}$$

where:

UDRT = upper delay range threshold (default 90ms)

FT = feedback time interval (default 50ms)

w = number of repeated timeouts (w=0 initially, w++ on each timeout, and reset to 0 when a message is received)

beginning when the last message (of any type) was successfully received at the sender:

Then the offered load SHALL be decreased, following the same process as when the feedback indicates presence of one or more sequence number anomalies OR the delay range was above the upper threshold (as described above), with the same load rate adjustment algorithm variables in their current state. This means that rate reduction and congestion confirmation can result from a three-way OR that includes lost status feedback messages, sequence errors, or delay variation.

The RECOMMENDED initial value for w is 0, taking Round Trip Time (RTT) less than FT into account. A test with RTT longer than FT is a valid reason to increase the initial value of w appropriately. Variable w SHALL be incremented by 1 whenever the Lost Status Backoff timeout is exceeded. So with FT = 50ms and UDRT = 90ms, a status feedback message loss would be declared at 190ms following a successful message, again at 50ms after that (240ms total), and so on.

Also, if congestion is now confirmed for the first time by a Lost Status Backoff timeout, then the offered load rate is decreased by more than one rate (e.g., Rx-30). This one-time reduction is intended to compensate for the fast initial ramp-up. In all other cases, the offered load rate is only decreased by one (Rx-1).

[Appendix B](#) discusses compliance with the applicable mandatory requirements of [\[RFC8085\]](#), consistent with the goals of the IP-Layer Capacity Metric and Method, including the load rate adjustment algorithm described in this section.

8.2. Measurement Qualification or Verification

It is of course necessary to calibrate the equipment performing the IP-Layer Capacity measurement, to ensure that the expected capacity can be measured accurately, and that equipment choices (processing speed, interface bandwidth, etc.) are suitably matched to the measurement range.

When assessing a Maximum rate as the metric specifies, artificially high (optimistic) values might be measured until some buffer on the path is filled. Other causes include bursts of back-to-back packets with idle intervals delivered by a path, while the measurement interval (dt) is small and aligned with the bursts. The artificial values might result in an un-sustainable Maximum Capacity observed when the method of measurement is searching for the Maximum, and that would not do. This situation is different from the bi-modal service rates (discussed under Reporting), which are characterized by a multi-second duration (much longer than the measured RTT) and repeatable behavior.

There are many ways that the Method of Measurement could handle this false-max issue. The default value for measurement of singletons (dt = 1 second) has proven to be of practical value during tests of this method, allows the bimodal service rates to be characterized, and it has an obvious alignment with the reporting units (Mbps).

Another approach comes from [Section 24 of RFC 2544](#)[\[RFC2544\]](#) and its discussion of Trial duration, where relatively short trials conducted as part of the search are followed by longer trials to make the final determination. In the production network, measurements of Singletons and Samples (the terms for trials and tests of Lab Benchmarking) must be limited in duration because they may be service-affecting. But there is sufficient value in repeating a Sample with a fixed sending rate determined by the previous search for the Maximum IP-Layer Capacity, to qualify the result in terms of the other performance metrics measured at the same time.

A qualification measurement for the search result is a subsequent measurement, sending at a fixed 99.x % of the Maximum IP-Layer Capacity for I, or an indefinite period. The same Maximum Capacity Metric is applied, and the Qualification for the result is a Sample without packet loss or a growing minimum delay trend in subsequent singletons (or each dt of the measurement interval, I). Samples exhibiting losses or increasing queue occupation require a repeated search and/or test at reduced fixed sender rate for qualification.

Here, as with any Active Capacity test, the test duration must be kept short. 10 second tests for each direction of transmission are common today. The default measurement interval specified here is I = 10 seconds. The combination of a fast and congestion-aware search method and user-network coordination make a unique contribution to production testing. The Maximum IP Capacity metric and method for assessing performance is very different from classic [\[RFC2544\]](#) Throughput metric and methods : it uses near-real-time load adjustments that are sensitive to loss and delay, similar to other congestion control algorithms used on the Internet every day, along with limited duration. On the other hand, [\[RFC2544\]](#) Throughput measurements can produce sustained overload conditions for extended periods of time. Individual trials in a test governed by a binary search can last 60 seconds for each step, and the final confirmation trial may be even longer. This is very different from "normal" traffic levels, but overload conditions are not a concern in the isolated test environment. The concerns raised in [\[RFC6815\]](#) were that [\[RFC2544\]](#) methods would be let loose on production networks, and instead the authors challenged the standards community to develop metrics and methods like those described in this memo.

8.3. Measurement Considerations

In general, the wide-spread measurements that this memo encourages will encounter wide-spread behaviors. The bimodal IP Capacity behaviors already discussed in [Section 6.6](#) are good examples.

In general, it is RECOMMENDED to locate test endpoints as close to the intended measured link(s) as practical (this is not always possible for reasons of scale; there is a limit on number of test endpoints coming from many perspectives, management and measurement traffic for example). The testing operator MUST set a value for the MaxHops parameter, based on the expected path length. This parameter can keep measurement traffic from straying too far beyond the intended path.

The path measured may be state-full based on many factors, and the Parameter "Time of day" when a test starts may not be enough information. Repeatable testing may require the time from the beginning of a measured flow, and how the flow is constructed including how much traffic has already been sent on that flow when a state-change is observed, because the state-change may be based on time or bytes sent or both. Both load packets and status feedback messages MUST contain sequence numbers, which helps with measurements based on those packets.

Many different traffic shapers and on-demand access technologies may be encountered, as anticipated in [[RFC7312](#)], and play a key role in measurement results. Methods MUST be prepared to provide a short preamble transmission to activate on-demand access, and to discard the preamble from subsequent test results.

Conditions which might be encountered during measurement, where packet losses may occur independently from the measurement sending rate:

1. Congestion of an interconnection or backbone interface may appear as packet losses distributed over time in the test stream, due to much higher rate interfaces in the backbone.
2. Packet loss due to use of Random Early Detection (RED) or other active queue management may or may not affect the measurement flow if competing background traffic (other flows) are simultaneously present.
3. There may be only small delay variation independent of sending rate under these conditions, too.

4. Persistent competing traffic on measurement paths that include shared transmission media may cause random packet losses in the test stream.

It is possible to mitigate these conditions using the flexibility of the load-rate adjusting algorithm described in [Section 8.1](#) above (tuning specific parameters).

If the measurement flow burst duration happens to be on the order of or smaller than the burst size of a shaper or a policer in the path, then the line rate might be measured rather than the bandwidth limit imposed by the shaper or policer. If this condition is suspected, alternate configurations SHOULD be used.

In general, results depend on the sending stream characteristics; the measurement community has known this for a long time, and needs to keep it front of mind. Although the default is a single flow ($F=1$) for testing, use of multiple flows may be advantageous for the following reasons:

1. the test hosts may be able to create higher load than with a single flow, or parallel test hosts may be used to generate 1 flow each.
2. there may be link aggregation present (flow-based load balancing) and multiple flows are needed to occupy each member of the aggregate.
3. access policies may limit the IP-Layer Capacity depending on the Type-P of packets, possibly reserving capacity for various stream types.

Each flow would be controlled using its own implementation of the load rate adjustment (search) algorithm.

As testing continues, implementers should expect some evolution in the methods. The ITU-T has published a Supplement (60) to the Y-series of Recommendations, "Interpreting ITU-T Y.1540 Maximum IP-Layer Capacity measurements", [[Y.Sup60](#)], which is the result of continued testing with the metric, and those results have improved the method described here.

[8.4.](#) Running Code

This section is for the benefit of the Document Shepherd's form, and will be deleted prior to final review.

Much of the development of the method and comparisons with existing methods conducted at IETF Hackathons and elsewhere have been based on the example udpst Linux measurement tool (which is a working reference for further development) [[udpst](#)]. The current project:

- o is a utility that can function as a client or server daemon
- o requires a successful client-initiated setup handshake between cooperating hosts and allows firewalls to control inbound unsolicited UDP which either go to a control port [expected and w/ authentication] or to ephemeral ports that are only created as needed. Firewalls protecting each host can both continue to do their job normally. This aspect is similar to many other test utilities available.
- o is written in C, and built with gcc (release 9.3) and its standard run-time libraries
- o allows configuration of most of the parameters described in Sections [4](#) and [7](#).
- o supports IPv4 and IPv6 address families.
- o supports IP-Layer packet marking.

[9.](#) Reporting Formats

The singleton IP-Layer Capacity results SHOULD be accompanied by the context under which they were measured.

- o timestamp (especially the time when the maximum was observed in dtn)
- o Source and Destination (by IP or other meaningful ID)
- o other inner parameters of the test case ([Section 4](#))
- o outer parameters, such as "test conducted in motion" or other factors belonging to the context of the measurement
- o result validity (indicating cases where the process was somehow interrupted or the attempt failed)
- o a field where unusual circumstances could be documented, and another one for "ignore/mask out" purposes in further processing

The Maximum IP-Layer Capacity results SHOULD be reported in the format of a table with a row for each of the test Phases and Number

of Flows. There SHOULD be columns for the phases with number of flows, and for the resultant Maximum IP-Layer Capacity results for the aggregate and each flow tested.

As mentioned in [Section 6.6](#), bi-modal (or multi-modal) maxima SHALL be reported for each mode separately.

Phase, # Flows	Maximum IP-Layer Capacity, Mbps	Loss Ratio	RTT min, max, msec
Search,1	967.31	0.0002	30, 58
Verify,1	966.00	0.0000	30, 38

Maximum IP-layer Capacity Results

Static and configuration parameters:

The sub-interval time, dt, MUST accompany a report of Maximum IP-Layer Capacity results, and the remaining Parameters from [Section 4](#), General Parameters.

The PM list metrics corresponding to the sub-interval where the Maximum Capacity occurred MUST accompany a report of Maximum IP-Layer Capacity results, for each test phase.

The IP-Layer Sender Bit rate results SHOULD be reported in the format of a table with a row for each of the test phases, sub-intervals (st) and number of flows. There SHOULD be columns for the phases with number of flows, and for the resultant IP-Layer Sender Bit rate results for the aggregate and each flow tested.

Phase, Flow or Aggregate	st, sec	Sender Bitrate, Mbps
Search,1	0.00 - 0.05	345
Search,2	0.00 - 0.05	289
Search,Agg	0.00 - 0.05	634

IP-layer Sender Bit Rate Results

Static and configuration parameters:

The subinterval time, *st*, MUST accompany a report of Sender IP-Layer Bit Rate results.

Also, the values of the remaining Parameters from [Section 4](#), General Parameters, MUST be reported.

9.1. Configuration and Reporting Data Formats

As a part of the multi-Standards Development Organization (SDO) harmonization of this metric and method of measurement, one of the areas where the Broadband Forum (BBF) contributed its expertise was in the definition of an information model and data model for configuration and reporting. These models are consistent with the metric parameters and default values specified as lists in this memo. [\[TR-471\]](#) provides the Information model that was used to prepare a full data model in related BBF work. The BBF has also carefully considered topics within its purview, such as placement of measurement systems within the access architecture. For example, timestamp resolution requirements that influence the choice of the test protocol are provided in Table 2 of [\[TR-471\]](#).

10. Security Considerations

Active metrics and measurements have a long history of security considerations. The security considerations that apply to any active measurement of live paths are relevant here. See [\[RFC4656\]](#) and [\[RFC5357\]](#).

When considering privacy of those involved in measurement or those whose traffic is measured, the sensitive information available to potential observers is greatly reduced when using active techniques which are within this scope of work. Passive observations of user traffic for measurement purposes raise many privacy issues. We refer the reader to the privacy considerations described in the Large Scale Measurement of Broadband Performance (LMAP) Framework [\[RFC7594\]](#), which covers active and passive techniques.

There are some new considerations for Capacity measurement as described in this memo.

1. Cooperating Source and Destination hosts and agreements to test the path between the hosts are REQUIRED. Hosts perform in either the Src or Dst roles.
2. It is REQUIRED to have a user client-initiated setup handshake between cooperating hosts that allows firewalls to control inbound unsolicited UDP traffic which either goes to a control port [expected and w/authentication] or to ephemeral ports that

are only created as needed. Firewalls protecting each host can both continue to do their job normally.

3. Client-server authentication and integrity protection for feedback messages conveying measurements is RECOMMENDED.
4. Hosts MUST limit the number of simultaneous tests to avoid resource exhaustion and inaccurate results.
5. Senders MUST be rate-limited. This can be accomplished using a pre-built table defining all the offered load rates that will be supported ([Section 8.1](#)). The recommended load-control search algorithm results in "ramp-up" from the lowest rate in the table.
6. Service subscribers with limited data volumes who conduct extensive capacity testing might experience the effects of Service Provider controls on their service. Testing with the Service Provider's measurement hosts SHOULD be limited in frequency and/or overall volume of test traffic (for example, the range of I duration values SHOULD be limited).

The exact specification of these features is left for the future protocol development.

[11.](#) IANA Considerations

This memo makes no requests of IANA.

[12.](#) Acknowledgments

Thanks to Joachim Fabini, Matt Mathis, J. Ignacio Alvarez-Hamelin, Wolfgang Balzer, Frank Brockners, Greg Mirsky, Martin Duke, Murray Kucherawy, and Benjamin Kaduk for their extensive comments on the memo and related topics.

[13.](#) [Appendix A](#) - Load Rate Adjustment Pseudo Code

The following is a pseudo-code implementation of the algorithm described in [Section 8.1](#).


```
Rx = 0 # The current sending rate (equivalent to a row of the table)
seqErr = 0 # Measured count of any of Loss or Reordering impairments
delay = 0 # Measured Range of Round Trip Delay, RTD, ms
lowThresh = 30 # Low threshold on the Range of RTD, ms
upperThresh = 90 # Upper threshold on the Range of RTD, ms
hSpeedTresh = 1 Gbps # Threshold for transition between sending rate step
  sizes (such as 1 Mbps and 100 Mbps)
slowAdjCount = 0 # Measured Number of consecutive status reports
  indicating loss and/or delay variation above upperThresh
slowAdjThresh = 2 # Threshold on slowAdjCount used to infer congestion.
  Use values >1 to avoid misinterpreting transient loss
highSpeedDelta = 10 # The number of rows to move in a single adjustment
  when initially increasing offered load (to ramp-up quickly)
maxLoadRates = 2000 # Maximum table index (rows)
```

```
if ( seqErr == 0 && delay < lowThresh ) {
    if ( Rx < hSpeedTresh && slowAdjCount < slowAdjThresh ) {
        Rx += highSpeedDelta;
        slowAdjCount = 0;
    } else {
        if ( Rx < maxLoadRates - 1 )
            Rx++;
    }
} else if ( seqErr > 0 || delay > upperThresh ) {
    slowAdjCount++;
    if ( Rx < hSpeedTresh && slowAdjCount == slowAdjThresh ) {
        if ( Rx > highSpeedDelta * 3 )
            Rx -= highSpeedDelta * 3;
        else
            Rx = 0;
    } else {
        if ( Rx > 0 )
            Rx--;
    }
}
```

14. Appendix B - RFC 8085 UDP Guidelines Check

The BCP on UDP usage guidelines [[RFC8085](#)] focuses primarily on congestion control in [section 3.1](#). The Guidelines appear in mandatory (MUST) and recommendation (SHOULD) categories.

14.1. Assessment of Mandatory Requirements

The mandatory requirements in [Section 3 of \[RFC8085\]](#) include:

Internet paths can have widely varying characteristics, ... Consequently, applications that may be used on the Internet MUST NOT make assumptions about specific path characteristics. They MUST instead use mechanisms that let them operate safely under very different path conditions. Typically, this requires conservatively probing the current conditions of the Internet path they communicate over to establish a transmission behavior that it can sustain and that is reasonably fair to other traffic sharing the path.

The purpose of the load rate adjustment algorithm in [Section 8.1](#) is to probe the network and enable Maximum IP-Layer Capacity measurements with as few assumptions about the measured path as possible, and within the range application described in [Section 2](#). The degree of probing conservatism is in tension with the need to minimize both the traffic dedicated to testing (especially with Gigabit rate measurements) and the duration of the test (which is one contributing factor to the overall algorithm fairness).

The text of [Section 3 of \[RFC8085\]](#) goes on to recommend alternatives to UDP to meet the mandatory requirements, but none are suitable for the scope and purpose of the metrics and methods in this memo. In fact, ad hoc TCP-based methods fail to achieve the measurement accuracy repeatedly proven in comparison measurements with the running code [\[LS-SG12-A\]](#) [\[LS-SG12-B\]](#) [\[Y.Sup60\]](#). Also, the UDP aspect of these methods is present primarily to support modern Internet transmission where a transport protocol is required [\[copycat\]](#); the metric is based on the IP-Layer and UDP allows simple correlation to the IP-Layer.

[Section 3.1.1 of \[RFC8085\]](#) discusses protocol timer guidelines:

Latency samples MUST NOT be derived from ambiguous transactions. The canonical example is in a protocol that retransmits data, but subsequently cannot determine which copy is being acknowledged.

Both load packets and status feedback messages MUST contain sequence numbers, which helps with measurements based on those packets, and there are no retransmissions needed.

When a latency estimate is used to arm a timer that provides loss detection -- with or without retransmission -- expiry of the timer MUST be interpreted as an indication of congestion in the network, causing the sending rate to be adapted to a safe conservative rate...

The method described in this memo uses timers for sending rate backoff when status feedback messages are lost (Lost Status Backoff

timeout), and for stopping a test when connectivity is lost for an longer interval (Feedback message or load packet timeouts).

There is no specific benefit foreseen by using Explicit Congestion Notification (ECN) in this memo.

[Section 3.2 of \[RFC8085\]](#) discusses message size guidelines:

To determine an appropriate UDP payload size, applications MUST subtract the size of the IP header (which includes any IPv4 optional headers or IPv6 extension headers) as well as the length of the UDP header (8 bytes) from the PMTU size.

The method uses a sending rate table with a maximum UDP payload size that anticipates significant header overhead and avoids fragmentation.

[Section 3.3 of \[RFC8085\]](#) provides reliability guidelines:

Applications that do require reliable message delivery MUST implement an appropriate mechanism themselves.

The IP-Layer Capacity Metric and Method do not require reliable delivery.

Applications that require ordered delivery MUST reestablish datagram ordering themselves.

The IP-Layer Capacity Metric and Method does not need to reestablish packet order; it is preferred to measure packet reordering if it occurs [\[RFC4737\]](#).

[14.2. Assessment of Recommendations](#)

The load rate adjustment algorithm's goal is to determine the Maximum IP-Layer Capacity in the context of an infrequent, diagnostic, short term measurement. This goal is a global exception to many [\[RFC8085\]](#) SHOULD-level requirements, of which many are intended for long-lived flows that must coexist with other traffic in more-or-less fair way. However, the algorithm (as specified in [Section 8.1](#) and [Appendix A](#) above) reacts to indications of congestion in clearly defined ways.

A specific recommendation is provided as an example. [Section 3.1.5 of \[RFC8085\]](#) on implications of RTT and Loss Measurements on Congestion Control says:

A congestion control designed for UDP SHOULD respond as quickly as possible when it experiences congestion, and it SHOULD take into

	Y? RFC 8085 Recommendation	Section
Yes	MUST tolerate a wide range of Internet path conditions	3
NA	SHOULD use a full-featured transport (e.g., TCP)	
Yes	SHOULD control rate of transmission	3.1
NA	SHOULD perform congestion control over all traffic	
	for bulk transfers,	3.1.2
NA	SHOULD consider implementing TFRC	
NA	else, SHOULD in other ways use bandwidth similar to TCP	

	for non-bulk transfers,	3.1.3
NA	SHOULD measure RTT and transmit max. 1 datagram/RTT	3.1.1
NA	else, SHOULD send at most 1 datagram every 3 seconds	
NA	SHOULD back-off retransmission timers following loss	
Yes	SHOULD provide mechanisms to regulate the bursts of transmission	3.1.6
NA	MAY implement ECN; a specific set of application mechanisms are REQUIRED if ECN is used.	3.1.7
Yes	for DiffServ, SHOULD NOT rely on implementation of PHBs	3.1.8
Yes	for QoS-enabled paths, MAY choose not to use CC	3.1.9
Yes	SHOULD NOT rely solely on QoS for their capacity	3.1.10
	non-CC controlled flows SHOULD implement a transport circuit breaker	
	MAY implement a circuit breaker for other applications	
	for tunnels carrying IP traffic,	3.1.11
NA	SHOULD NOT perform congestion control	
NA	MUST correctly process the IP ECN field	
	for non-IP tunnels or rate not determined by traffic,	
NA	SHOULD perform CC or use circuit breaker	3.1.11
NA	SHOULD restrict types of traffic transported by the tunnel	
Yes	SHOULD NOT send datagrams that exceed the PMTU, i.e.,	3.2
Yes	SHOULD discover PMTU or send datagrams < minimum PMTU;	
NA	Specific application mechanisms are REQUIRED if PLPMTUD is used.	
Yes	SHOULD handle datagram loss, duplication, reordering	3.3
NA	SHOULD be robust to delivery delays up to 2 minutes	
Yes	SHOULD enable IPv4 UDP checksum	3.4
Yes	SHOULD enable IPv6 UDP checksum; Specific application mechanisms are REQUIRED if a zero IPv6 UDP checksum is used.	3.4.1
NA	SHOULD provide protection from off-path attacks	5.1
	else, MAY use UDP-Lite with suitable checksum coverage	3.4.2
NA	SHOULD NOT always send middlebox keep-alive messages	3.5
NA	MAY use keep-alives when needed (min. interval 15 sec)	

Yes	Applications specified for use in limited use (or controlled environments) SHOULD identify equivalent mechanisms and describe their use case.	3.6
NA	Bulk-multicast apps SHOULD implement congestion control	4.1.1
NA	Low volume multicast apps SHOULD implement congestion control	4.1.2
NA	Multicast apps SHOULD use a safe PMTU	4.2
Yes	SHOULD avoid using multiple ports	5.1.2
Yes	MUST check received IP source address	
NA	SHOULD validate payload in ICMP messages	5.2
Yes	SHOULD use a randomized source port or equivalent technique, and, for client/server applications, SHOULD send responses from source address matching request	6
	5.1	
NA	SHOULD use standard IETF security protocols when needed	6
+-----+-----+		

15. References

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